When a VoIP call is established between TAOS unit-173 and TAOS unit-196, call data is transmitted across the packet network using the IP addresses of the Ethernet card ports on these two TAOS units (for example, 208.168.25.173/24 and 208.168.25.196).

For TAOS unit-196, the IP addresses in Table 2-7 are used as logical network gateways to send/receive VoIP call data across the IP network and are associated with network ports on this TAOS unit system's Ethernet-3 cards.

Table 2-7. Host route IP address table for TAOS unit-*196*

System IP address	Ethernet interfaces	IP address
192.168.35.196	{{000}0}*	192.168.35.196/24
	{{131}0}	208.168.25.196/24
	{{141}0}	208.168.15.196/24

* This is the soft IP address by which the TAOS unit is known to the network.

TAOS unit-196 uses the IP addresses in Table 2-8 to define the destination IP address it uses for sending VoIP call data across the network to TAOS unit-173.

Table 2-8. Host route IP Route table for TAOS unit-*196*

dest-address parameter*	gateway-address parameter†
192.168.35.173/32	208.168.25.173
192.168.35.173/32	208.168.15.173

* Using the 32-bit subnet address fully qualifies the IP address, making this a host route.
† This is the Ethernet port IP address of the destination TAOS unit.

Configuring network routes

Although configuring network routes requires smarter logic in the IP network, it's much simpler to configure the TAOS unit. The general idea here is to have only one default route for each Ethernet slot card in each TAOS unit, regardless of how many destination gateways or end points exist. For this to work the following rules apply:

- The value of `system-ip-addr` assigned each TAOS unit must:
  - Be unique.
  - Be on a logically different subnet than any IP address assigned to any Ethernet card in any MultiVoice network.
- No two Ethernet slot card IP addresses may reside on the same logical subnet.
- Routers in the IP network must also have equal-cost multipath routes, so that VoIP packets will get routed to the IP interface on an Ethernet slot card on the destination APX in an equal load-balanced fashion.

For example, the following tables illustrate the IP addressing scheme you use to allow TAOS unit-173 and TAOS unit-196 to connect VoIP calls between them, across an IP network which supports equal-cost multipath routing, using two equal-cost routes.

The IP addresses in Table 2-9 are used as logical network gateways by the TAOS unit-173 to send/receive VoIP call data across the IP network. These are associated with network ports on the Ethernet-3 cards that provide the physical network connection to the packet network.

The value assigned to the `system-ip-addr` parameter is used as the network IP address for TAOS unit-173 itself. The MultiVoice administrator can set the value of the `system-ip-addr` parameter to either the soft IP address configured for the TAOS unit or the IP address of the Ethernet interface on the shelf controller. For this illustration, the `system-ip-addr` parameter for TAOS unit-173 uses the soft IP address.

Table 2-9. Network route IP address table for TAOS unit-*173*

Ethernet interfaces	**IP address***
System-Ip-Addr†	Anything
{ { 1 3 1 } 0 }	208.168.25.173/24
{ { 1 4 1 } 0 }	208.168.15.173/24
{ { any-shelf any-slot 0} 0 }	208.168.35.173/32‡

* This IP address can't match the subnet of any Ethernet card IP address or System-Ip-Addr assigned across the network.
† This IP address must be on a logically different subnet from any other Ethernet card IP address assigned across the network.
‡ This is the soft IP address associated with the TAOS unit itself. It will be used for mapping the host route back from the network router.

The IP addresses in Table 2-10 identify the router(s) that direct VoIP calls to/from the IP network for TAOS unit-173. These routers provide the first/last hop on the IP network for calls involving TAOS unit-173. Each network route is configured by creating an `ip-route` profile using the default value for the `dest-address` parameter (`dest-address=0.0.0.0/0`) and one of the IP addresses associated with a network gateway router, that provides the connection between TAOS unit-173 and the packet network, is assigned to the `gateway-address` parameter.

Table 2-10. Network route IP Route table for TAOS unit-*173*

dest-address parameter	**gateway-address parameter***
0.0.0.0	208.168.25.1
0.0.0.0	208.168.15.1

* This is the IP address of the network router used for connecting calls with destination TAOS units.

A network route ip-route profile would be similar to the following:

```
admin> list
[in IP-ROUTE/NorthAmerica1]
name* = NorthAmerica1
dest-address = 0.0.0.0/0
gateway-address = 208.168.25.1
```

The IP addresses in Table 2-11 are used by the network router to connect TAOS unit-173 with any destination MultiVoice Gateway or end point. These addresses point back from the network gateway router to TAOS unit-173. The network route method of configuration needs at least two equal-cost host routes pointing back to TAOS unit-173. These pointers should use the soft IP address created for TAOS unit-173 and one of the IP addresses associated with ports on the Ethernet-3 cards in TAOS unit-173.

Table 2-11. Network route IP Route table for network router

Destination IP address*	gateway address†
208.168.35.173/32	208.168.25.173
208.168.35.173/32	208.168.15.173

* Using the 32-bit subnet address fully qualifies the host IP address for the route back to TAOS unit-173. This should match the soft IP address defined for this TAOS unit.
† This is the IP Address of the Ethernet card on TAOS unit-173 used to point back to TAOS unit-173 from the network gateway router.

Any intermediate router between the network gateway router and the destination MultiVoice Gateway or end point should be configured to route VoIP calls meant for IP address 208.168.35.173 to the network gateway router. The router should then equally distribute the packets over the two Ethernet subnets pointing to TAOS unit-173.

The IP addresses in Table 2-12 are used as logical network gateways by TAOS unit-196 to send and receive VoIP call data across the IP network. These are associated with network ports on the Ethernet-3 slot cards that provide the physical network connection to the packet network.

The value assigned to the system-ip-addr parameter is used as the network IP address for TAOS unit-196 itself. The MultiVoice administrator can set the value of the system-ip-addr parameter to either the soft IP address configured for the TAOS unit or the IP address of the Ethernet interface on the shelf controller. For this illustration, the system-ip-addr parameter for TAOS unit-196 uses the soft IP address.

Table 2-12. Network route IP address table for TAOS unit-*196 (Page 1 of 2)*

Ethernet interfaces	IP address*
system-ip-addr†	Anything
{ { 1 3 1 } 0 }	208.168.25.196/24

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Table 2-12. Network route IP address table for TAOS unit-*196 (Page 2 of 2)*

Ethernet interfaces	IP address*
{ { 1 4 1 } 0 }	208.168.15.196/24
{ { any-shelf any-slot 0} 0 }	208.168.35.196/32‡

* This IP address can't match the subnet of any Ethernet card IP address or system-ip-addr assigned across the network.
† This IP address must be on a logically different subnet from any other Ethernet card IP address assigned across the network.
‡ This is the soft IP address associated with the TAOS unit itself. It will be used for mapping the host route back from the network router.

The IP addresses in Table 2-13 identify the router(s) that direct VoIP calls to/from the IP network for TAOS unit-196. These routers provide the first/last hop on the IP network for calls involving TAOS unit-196. Each network route is configured by creating an ip-route profile using the default value for the dest-address parameter (dest-address=0.0.0.0/0) and one of the IP addresses associated with a network gateway router, that provides the connection between TAOS unit-196 and the packet network, is assigned to the gateway-address parameter. This router provides the first/last hop on the IP network for calls involving TAOS unit-196.

Table 2-13. Network route IP Route table for TAOS unit-*196*

dest-address parameter	gateway-address parameter*
0.0.0.0	208.168.45.1
0.0.0.0	208.168.55.1

* This is the IP address of the network router used for connecting calls with destination TAOS units.

A network route ip-route profile for TAOS unit-196 would be similar to the following:

```
admin> list
[in IP-ROUTE/NorthAmerica2]
name* = NorthAmerica2
dest-address = 0.0.0.0/0
gateway-address = 208.168.55.1
```

The IP addresses in Table 2-14 are used by the network router to connect TAOS unit-196 with any destination MultiVoice Gateway or end point. These addresses are used to point back from the router to TAOS unit-196, using its soft IP address.

Table 2-14. Network route IP Route table for network router (Page 1 of 2)

Destination IP address*	Gateway address†
208.168.65.196/32	208.168.45.196

Table 2-14. Network route IP Route table for network router (Page 2 of 2)

Destination IP address*	Gateway address†
208.168.65.196/32	208.168.55.196

* Using the 32-bit subnet address fully qualifies the host IP address for the route back to TAOS unit-196. This should match the soft IP address defined for this TAOS unit.
† This is the IP Address of the Ethernet card on TAOS unit-196 used to point back to TAOS unit-196 from the network gateway router.

Creating static routes

Both the host route method and network route method utilize static routes to send VoIP calls across the packet network. To create static routes for either host or network routes:

1 Assign an IP address to the first port on the Ethernet card in shelf 1, slot 3 of TAOS unit-173:

```
admin> read ip-inter { { 1 3 1} 0 }
IP-INTERFACE/{ { shelf-1 slot-3 1 } 0 } read
admin> set ip-address = 208.168.25.173/24
admin> write
IP-INTERFACE/{ { shelf-1 slot-3 1 } 0 } written
```

2 After assigning an IP address to at least one Ethernet port on each TAOS unit in the VoIP network, assign the appropriate IP address to the Gateway-Address parameter which points to an Ethernet port on a distant TAOS unit, to configure a static route between them. For example:

```
admin> new ip-route zone1
IP-ROUTE/zone1 read
admin> set dest-address = 197.88.10.2/32
admin> set gateway-address = 201.10.10.1
admin> write
IP-ROUTE/zone1 written
```


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For each of the VoIP static routes, the following values are applied to these parameters in the `ip-route` profile:

Table 2-15. Static route parameters

Parameter	Specifies	
	Host Route	**Network Route**
dest-address	This is the fully qualified IP address assigned to the `system-ip-addr` parameter in the `ip-global` profile of a destination gateway or end point. This IP address must include the host port identifier (such as, 197.88.10.2/32).	This is always 0.0.0.0. The network router completes the connection to the appropriate IP address on the destination gateway or end point.
gateway-address	This IP address identifies a port on one of the Ethernet slot cards installed on the destination gateway/end point.	This IP address identifies the network router used to contact an TAOS unit which resides on another subnet of the network.

Using multipath routes for VoIP

Using multipath static routes for VoIP traffic across the IP network distributes VoIP traffic across the aggregated bandwidth of multiple Ethernet interfaces. This increases the number of simultaneous calls that can processed between destinations and reduces call wait times and the number of rejected calls. Each multipath route requires static routes that meet the following criteria:

- Static routes have the same destination address and subnet mask, but different gateway addresses (such as different Ethernet port IP addresses, on different Ethernet slot cards).
- The routes have the same route metric.
- The routes have the same route preference.

If more than one VoIP call comes in for the same destination TAOS unit, the local TAOS unit will check all routes within a multipath route, on a call-by-call basis, and selects the first available route it finds for each call. (For additional information on IP routing, see the *APX 8000/MAX TNT WAN, Routing, and Tunneling Configuration Guide.*)

Configuring routes for IPDC VoIP call processing

Both APX and MAX TNT support IPDC 0.12-controlled VoIP calls over InterMachine trunks (IMTs) for SS7 calls originating from the PSTN. IPDC message tags define all the VoIP parameter values used for processing VoIP calls, overriding the default `voip` profile (`voip { 0 0 }`). When initiating an IPDC VoIP call, tag values determine voice

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encoding type, packet loading, IP and RPT ports, etc. The default `voip` profile (`voip { 0 0 }`) value applies only when a specific parameter value is not specified by an IPDC message.

Routing is controlled from the signaling gateway (for example, Lucent Softswitch), which issues instructions for establishing the TDM/IP/RTP connection between the TAOS units involved in the call. Once this connection is established, the TAOS units pass RTP packets, bidirectionally, across the packet network.

Configuring IP routing for IPDC call processing

There are two methods for implementing IP routing for IPDC VoIP call processing:

- Using RTP listen IP addresses selected by the TAOS unit
- Using RTP listen IP addresses selected by the signaling gateway

When the TAOS unit selects its own listen IP address using RMCP/AMCP messages (see "Using RMCP/AMCP messages to route VoIP calls" on page 2-39), VoIP packet routing and configuration is managed in the same way as H.323 VoIP call processing. See "Packet routing for H.323 VoIP calls" on page 2-29 for details.

When the signaling gateway is configured to perform equal-cost routing across multiple IP addresses (each associated with the Ethernet slot cards in the TAOS unit), each Ethernet port must be assigned IP addresses residing on different logical subnets. In this instance, the signaling gateway determines what IP address to use for the call, with based weighting algorithms used by the media gateway controller application.

Packet routing for IPDC VoIP calls controlled from the signaling gateway

To load balance IPDC VoIP calls across Ethernet slot cards (have the signaling gateway allocate IP addresses in an equal fashion) all Ethernet slot card IP addresses must reside on a different logical subnet within the same TAOS unit. Since routing is controlled by the media gateway controller application on the signaling gateway, this is the only configuration required on the TAOS unit. For this to work, the following rules apply:

- The value of the `system-ip-addr` parameter for each TAOS unit must:
  - Be unique
  - Be on a logically different subnet from any IP address assigned to any Ethernet card in any MultiVoice network
- No two Ethernet slot card IP addresses assigned to the same TAOS unit may reside on the same logical subnet
- The TAOS unit's IP address must be assigned to SS7-signal packets

The Ethernet IP address configuration, illustrated by Table 2-16, is an example of how IP addresses should be assigned to Ethernet cards to allow equal-cost routing to be done by the signaling gateway.

Table 2-16. Ethernet IP address table for TAOS unit-173

System IP address	Ethernet interfaces	IP address
192.168.35.173	{ { 0 0 0 } 0 }*	192.168.35.173/24
	{ { 1 3 1 } 0 }	208.168.25.173/24
	{ { 1 4 1 } 0 }	208.168.15.173/24
	{ { 1 5 1 } 0 }	208.168.5.173/24

* This is the soft IP address assigned to the TAOS unit.

When VoIP data is passed across the packet network between two TAOS units, the source address contained in the SS7 signaling transport packets is used to establish the return path for VoIP call data sent back to the originating TAOS unit. This source address is the IP address where intermediate network routers send data in response to SS7 VoIP data transmissions. To achieve equal-cost routing for IPDC VoIP calls, that source address must be the IP address (that is, the value assigned to the `system-ip-addr` parameter) of the originating TAOS unit.

For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24
admin> read ip-global
IP-GLOBAL read
admin> set system-ip-addr = 1.1.1.1
admin> write
IP-GLOBAL written
```

In addition, you must make sure that VoIP calls can always find a route to the next-hop MultiVoice Gateway on the path to the destination MultiVoice Gateway. The route can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so that RTP packets are never dropped because of lack of routing information. For example, the following commands configure a default route named VoIP to a next-hop MultiVoice Gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read
admin> set gateway = 2.2.2.2/24
admin> write
IP-ROUTE/VoIP written
```

The IP address of the TAOS unit is assigned to the SS7 signal packets by setting the value of the `use-system-ip-address-as-source` parameter, in the **`ss7-system`** profile, to yes. This inserts the IP address of the originating TAOS unit in the SS7 signaling

transport packets before transmission across the packet network. See "Using the ss7-gateway profile" on page 2-46.

Using RMCP/AMCP messages to route VoIP calls

The request modify call parameters (RMCP) and accept confirm call parameters (AMCP) messages are used to modify parameters for RCCP/ACCP controlled (VoIP) calls. The messages can be used to modify the following parameters:

- VoIP encoding type (G.711, G.729, and so forth) with Tag 0x70. Note that TAOS also supports G.723 (5.4 Kbps) encoding for SS7 VOIP calls. Following are the supported values for VOIP encoding:

Encoding type	Value
G.711 µ-law	0x00
G.723	0x04
G.711 a-law	0x08
G.729	0x12

- Packet Loading (frames/packet) with Tag 0x73. Values depend on VoIP encoding type.
- Destination Port Type with Tag 0x65. Note that for IPDC 0.12 VoIP calls, the only supported values for Source (0x65) and Dest Port (0x66) Type tags are SCN (0x00) and RTP (0x01) respectively.
- Listen IP address with Tag 0x5D.
- Listen RTP port with Tag 0x5E.
- Send IP address with Tag 0x5F.
- Send RTP port with Tag 0x60.

The table below shows the tags supported for the RMCP message:

Tag	Parameter Description	R / O status
0x65	Source port type (PSTN only)	Required
0x07	Source module number	Required
0x0D	Source line number	Required
0x15	Source channel number	Required
0x66	Destination port type (RTP only)	Required
0x70	VoIP encoding type (new G.723 value supported)	Optional
0x73	Packet loading (value depends on VOIP encoding type)	Optional
0x5D	Destination listen IP address (see Note below)	Optional
0x5E	Destination listen RTP port number (see Note below)	Optional
0x5F	Destination send IP address (see Note below)	Optional
0x60	Destination send RTP port number (see Note below)	Optional

Note The last four tags in the table are required if values are nonzero. In addition, if an IP address tag is present, the matching port tag must also be present.

This requirement also applies to the same tags in ACMP messages listed in the table below. RCCP and ACCP messages have been modified to use the same requirements regarding these tags.

The table below shows the tags supported for the AMCP message:

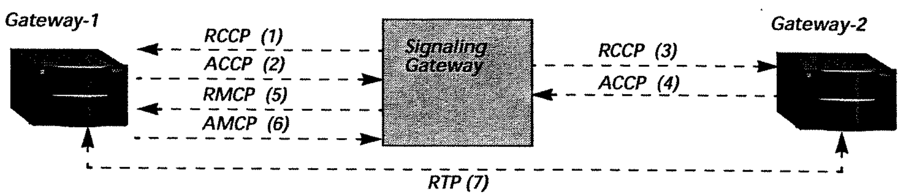
Tag	Parameter Description	R / O status
0x65	Source port type (PSTN only)	Required
0x07	Source module number	Required
0x0D	Source line number	Required
0x15	Source channel number	Required
0x66	Destination port type (RTP only)	Required
0x70	VOIP encoding type (new G.723 value supported)	Required
0x73	Packet loading (value depends on VOIP encoding type)	Required
0x5D	Destination listen IP address	Optional
0x5E	Destination listen RTP port number	Optional
0x5F	Destination send IP address	Optional
0x60	Destination send RTP port number	Optional

Tags 0x70 and 0x73 are required in ACMP messages because RMCP also queries the information for a VoIP call.

Send IP address and Send RTP port tags

With RMCP support of Tags 0x5F and 0x60, the TAOS unit can allocate its own listen IP addresses and RTP ports. The exchanges used in this process are shown in Figure 2-1:

Figure 2-1. Example IPDC message exchanges



IPDC messages to establish RTP listen addresses and ports are exchanged as follows:

1. The signaling gateway sends an RCCP message to Gateway-1, in which the RTP port [n] is either not specified or is 0, but with no IP address or RTP port tags.
2. Gateway-1 returns its RTP listen IP address and RTP port to the signaling gateway in an ACCP message, using tags 0x5D and 0x5E.
3. The signaling gateway sends an RCCP message to Gateway-2, in which the Destination listen IP address and Destination listen RTP port number obtained from Gateway-1 are specified.
4. Gateway-2 returns its RTP listen IP address and RTP port to the signaling gateway in an ACCP message, using tags 0x5D and 0x5E.
5. The signaling gateway sends an RMCP message to Gateway-1, in which the Destination listen IP address and Destination listen RTP port number obtained from Gateway-2 are specified.
6. Gateway-1 returns an AMCP message to the signaling gateway.
7. RTP communication commences between the Gateway-1 and Gateway-2.

Related routing issues

For all VoIP calls, it is important to avoid routing RTP traffic through the TAOS unit's shelf- controller. For that reason, when allowing the TAOS unit to allocate its own RTP address, you must set the system-ip-addr parameter in the ip-global profile to an interface address other than the default zero address (which defaults to the shelf-controller Ethernet port). For example, the following commands set the system address to the address of a port on an Ethernet card in slot 12:

```
admin> get ip-interface { { 1 12 1 } 0} ip-address
[in IP-INTERFACE/{ { shelf-1 slot-12 1 } 0 }:ip-address]
ip-address = 1.1.1.1/24
admin> read ip-global
IP-GLOBAL read
admin> set system-ip-addr = 1.1.1.1
admin> write
IP-GLOBAL written
```

In addition, it is important that VoIP calls can always find a route to the next-hop MultiVoice Gateway on the path to the destination MultiVoice Gateway. The route

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can be learned dynamically or configured as a static route. Many sites choose to configure default routes for VoIP traffic, so RTP packets will never be dropped due to lack of routing information. For example, the following commands configure a default route named voip to a next-hop gateway at 2.2.2.2:

```
admin> new ip-route voip
IP-ROUTE/voip read
admin> set gateway = 2.2.2.2/24
admin> write
IP-ROUTE/voip written
```

Reporting IPDC VoIP call statistics

A TAOS unit operating as a network access server (NAS) can report VoIP call statistics in the output of the NAS messaging interface. IPDC VoIP call statistics are reported once a call is cleared. The source that originates call clearing can be either the signaling gateway or the TAOS unit.

Supported tags for reporting statistics

IPDC 0.12 statistics tags are reported when the signaling gateway or a TAOS unit clears calls under the following conditions:

- The access server initiates a call teardown using an RCR message
- The access server acknowledges a call teardown using an ACR message for packet-based calls

Table 2-17 shows statistic-related tags from IPCD 0.12 that are currently supported by the MultiVoice Gateway with their descriptions:

Table 2-17. Supported Statistics Tags (IPDC 0.12) (Page 1 of 2)

Tag	Description
0x91	Number of Real-Time Protocol (RTP) audio packets sent and received by the APX.
0x92	Number of RTP audio packets that failed to reach the APX, determined by missed sequence numbers.
0x93	Number of audio bytes in the RTP payload sent by the APX.
0x94	Number of audio bytes received in the RTP payload that failed to reach the APX. Because the number of bytes per packet is variable, this value can only be estimated based upon an average packet size multiplied by the number of non-received packets. The control server can estimate this value with the information supplied.
0x9D	Number of RTP audio packets received.
0x9E	Number of audio bytes received in the RTP payload.

Table 2-17. Supported Statistics Tags (IPDC 0.12) (Page 2 of 2)

Tag	Description
0xA3	Estimated interarrival jitter (in milliseconds), which is computed as follows: `J = J + (D - J)/16` where `D =	R (i) - R(i-1) - T |`, `R (i)` is the arrival time of the received packet `i`, and `T` is the theoretical difference of departure time between two consecutive packets at the source. For example, `T` is 5 ms for G711 1 frame per packet, `T` is 10 ms for G729 1 frame per packet, and `T` is 40 ms for G729 4 frames per packet.

Unsupported tags for reporting statistics

Table 2-18 shows statistics-related tags from IPDC 0.12 that the MultiVoice Gateway does not support:

Table 2-18. Unsupported Statistics Tags (IPDC 0.12)

Tag	Description
0x95	Number of signaling packets sent and received.
0x96	Number of signaling packets dropped.
0x97	Number of signaling bytes sent and received.
0x98	Number of signaling bytes dropped.
0x99	Estimated average latency.
0x9F	Number of signaling packets received.
0xA0	Number of signaling bytes received.

Call statistics reporting

IPDC 0.15 specifies that the statistics tags are optional, and they are reported in the following cases of call clearing.

- The packet statistics information should be included when the access server initiates a call teardown via RCR message.
- The packet statistics should be included for packet-based calls when the access server acknowledges a call teardown via ACR message.

The TAOS unit reports the statistics in the above two cases when the statistics are available.

ss7nmi debug-level command

The TAOS unit reports VoIP call statistics in the output of the ss7nmi debug-level command. When the command is entered with the -s option, the results displayed include the number of release channel request (RCR) and release channel completed (ACR) messages sent with and without VoIP call statistics, and the number of unknown SS7 VoIP messages. In the following example, new statistics reported for IPDC VoIP calls are shown in bold type:

```
admin> ss7nmi -s
SS7 NAS Messaging Interface (NMI) statistics:
        Initialized successfully:                  Yes
        Total number of internal errors:           0
        Level of diagnostics:                      0
Signaling Layer:
        Current link state:                        UP
        Last generated transaction ID:             182
        Timer T305 (RST1):                         1000 ticks - idle
        Number of protocol version errors:         0
        Number of 'message reject' received:       0
        Number of bad packets received:            0
        Number of unknown messages:                0
        Number of unknown SS7Voip messages:        0
        Number of resource conflicts:              0
        Number of release race conditions:         0
        Number of RCR with stats sent:             0
        Number of RCR without stats sent:          0
        Number of ACR with stats sent:             36076
        Number of ACR without stats sent:          0
Data Transport Layer:
        Number of link fail-overs:                 0
        Number of persistent errors:               0
        Last error:                                No Error
        Last error timestamp:                      [09/02/1999 00:00:00]
```

Verifying IP route configuration

To verify the IP route configuration, check the Ethernet port caches and IP caches for each IP interface defined for an TAOS unit.

Verifying VoIP port caches

After creating routes for VoIP packet processing, run the Ipportmap command on the TAOS unit to verify the routing between a port on the Ethernet card and a destination TAOS unit. This command must be entered while calls are in progress, as in the following example:

1 Open a session with an Ethernet card on the TAOS unit. For example, if the Ethernet card is in Shelf 1, slot 2 of your TAOS unit:

```
admin> open 1 2
ether-1/2>
```

2 With a call in progress, enter the ipportmap command to verify port mappings:

```
ether-1/2> ipportmap -m
Port  Proto   Addr               Shelf/Slot         Refcnt
1469  UDP     192.168.35.131/32    1/6/0/0          12503
```

When the ports are mapped properly, the output displays the following information in these fields:

Field	Output
Addr	This is the fully qualified destination IP address of the TAOS unit that is using the network port on this Ethernet card to route VoIP packets.
Shelf/Slot	This is the shelf and slot address of the DSP card on the local TAOS unit which processes VoIP packets sent and received across the network.
Refcnt	This is the total number of packets received from the distant TAOS unit. The Refcnt field continues to increment while the call is in progress.

Verifying VoIP route caches

Enter the ipcache command to verify internal packet routing between an Ethernet card and a MultiDSP slot card on the TAOS unit. This command must be entered while calls are in progress, as in the following example.

1 Open a session with a the MultiDSP slot card. For example, if the DSP card is in shelf 1, slot 6 of your TAOS unit:

```
admin> open 1 6
madd-1/6>
```

2 With a call in progress, enter the Ipcache command to verify that the proper IP route cache was created:

```
madd-1/6> ipcache cache
Hash Address        Gateway          Shelf/Slot Type    MTU  Switched
142  192.168.35.173 208.168.15.173     1/2      STATIC  1500 512
     Forward Stats: To Slots 256, To Shelf 1
     Mem Usage: Allocated 1k bytes
                Free block count 24
```

When packet routes have been properly cached on the DSP slot card, the output display the following information in these fields:

Field	Output
Address	This is the IP address of the far-end TAOS unit that is sending VoIP packets to the local TAOS unit.
Gateway	This is the IP address of the network router, or the Ethernet card on the far-end TAOS unit, which is used to establish the packet network connection to the far-end TAOS unit.

Field	Output
Shelf/Slot	This is the shelf and slot address of the Ethernet card on the local TAOS unit which is used to establish the packet network connection with the far-end TAOS unit.

Trunk configuration

Trunk configuration controls how network signals are processed by the TAOS unit for VoIP calls. For calls originating from SS7 networks, the TAOS unit is configured to let the signaling gateway (manage network signal processing. For calls processed using H.323, the TAOS unit is configured to detect and respond to call progress signals from a PBX/PSTN.

Interoperating with an SS7 signaling gateway using IPDC

For every TAOS unit that interoperates with a signaling gateway, you must configure the following:

- IP interface to the SS7 signaling gateway running IPDC.
- IPDC messaging interface (SS7 profile).
- T1 line settings for SS7.

For more details on configuring the SS7 interface see the *APX 8000/MAX TNT Physical Interface Configuration Guide.*

Using the ss7-gateway profile

The signaling gateway and TAOS unit communicate over a TCP/IP link. The messaging interface can be a single or dual TCP connection between the TAOS unit and the signaling gateway. When the messaging interface initializes, it opens TCP connections to the specified addresses and ports of the signaling gateway. The TAOS unit keeps the TCP connections open as long as the unit is up and the IPDC messaging interface is enabled. Following are the parameters (shown with default settings) for configuring the messaging interface:

```
[in SS7-GATEWAY]
enabled = no
control-protocol = ipdc-0.X
primary-ip-address = 0.0.0.0
primary-tcp-port = 0
secondary-ip-address = 0.0.0.0
secondary-tcp-port = 0
bay-id =
system-type = IASCTNT1B
transport-options = { 0 1000 3000 30000 7 6 }
use-system-ip-address-as-source = yes
```


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Parameter	Setting
enabled	Whether the interface is enabled or disabled the interface. When set to no (the default), the interface is disabled. When set to yes, the interface is enabled if the primary-ip-address and primary-tcp-port also have valid values. Changing the setting from yes to no closes the signaling links but does not disconnect active SS7 calls.
control-protocol	The interface control protocol used for communications between the TAOS unit and the signaling gateway. The specified protocol provides call control for setting up, tearing down, and managing calls between the PSTN and the TAOS unit. The TAOS unit must be licensed for the appropriate code for the required control protocol to communicate with the signaling gateway. For VoIP, the TAOS unit must be licensed for IPDC 0.12, and the value of Control-Protocol should be set to ipdc-0.X.
primary-ip-address primary-tcp-port	IP address and TCP port to use as the primary IPDC interface. These settings are required to enable the messaging interface.
secondary-ip-address secondary-tcp-port	IP address and TCP port to use as the secondary IPDC interface. Typically, the primary and secondary address and port configurations point to the two Ethernet interfaces of the signaling gateway.
bay-id	This is an alphanumeric string which identifies the TAOS unit to the media gateway controller application. The content of this field is sent by TAOS unit to the signaling gateway during the device registration process. This parameter is not used with IPDC.
system-type	A device identifier used by the signaling gateway to identify the TAOS unit. This parameter is used for registration purposes only. The content of this field must be recorded on the signaling gateway.
transport-options	This subprofile tunes SS7 L2 timers.
use-system-ip-address-as-source	This parameter assigns either the TAOS unit IP address or signaling gateway address as the source address for SS7 signaling transport packets. The value identifies the IP address of the destination where intermediate network routers should direct data in response to SS7 VoIP/data transmissions. When this parameter is set to yes, the default, SS7 signaling transport packets are assigned the TAOS unit's IP address as their source, or outgoing physical interface address. When this parameter is set to no, SS7 signaling transport packets are assigned the signaling gateway IP address.

Transport-options subprofile

The transport-options subprofile allows users to make occasional changes in the operation of SS7 L2 (level 2) timers. The level 2 portion of the message transfer part

(MTP Level 2) provides link-layer functionality. It ensures that the two end points of a signaling link can reliably exchange signaling messages. It incorporates such capabilities as error checking, flow control, and sequence checking.

These timers manage the wait/response intervals for these various singling link processes. Changing these values are useful when customers need to use nonstandard values during system integration and for fine-tuning of their network. This subprofile contains the following fields:

```
[in SS7-GATEWAY:transport-options]
device-id = 0
t1-duration = 1000
t2-duration = 3000
t3-duration = 30000
windows-size = 7
ack-threshold = 6
```

Parameter	Setting
device-id	The logical SS7 command control device where these values apply. This is currently not used.
t1-duration	The value of the ACK delay timer in milliseconds. This timer specifies the maximum delay for an acknowledgement when an I-frame is received. Default value is 1000 (1 second). The value must be less than T2 on the peer. Valid values range from 0-2147483647.
t2-duration	The value of the transmission time-out timer in milliseconds. This timer specifies how long this end point should wait for an acknowledgement. Default value is 3000 (3 seconds). The value must be greater than T1 on the peer. Valid values range from 0 - 2147483647.
t3-duration	Value of the persistent error timer in milliseconds. This timer specifies the maximum duration of attempts to reestablish a link before the transport layer flushes the data queues and sends an error indication up. Default value is 30000 (30 seconds). Valid values range from 0-2147483647.
window-size	The maximum number of sequentially numbered data packets that can be sent while pending acknowledgement at any given time. Default value is 7. Valid values range from 1-63.
ack-threshold	The threshold for triggering an acknowledgment while receiving data packets. As soon as the specified number of packets is received, an ACK is sent back regardless of the value set for the T1 timer. The value of this parameter may not be greater than the window size. Default value is 6. Valid values range from 1-63.

Configuring an IP interface to the signaling gateway

For information about configuring LAN and WAN IP interfaces, see the *APX 8000/ MAX TNT WAN, Routing, and Tunneling Configuration Guide.* That guide also describes standard methods you can use to isolate the interface, such as making the route

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private or applying a route filter to the interface, to be certain that only the SS7 messages cross the link between the TAOS unit and the signaling gateway.

Configuring T1 or E1 lines as SS7 data trunks

To configure T1/E1 lines for SS7, you must set the following parameters, shown with sample settings:

```
[in T1/{ shelf-1 slot-1 7 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing
[in E1/{ shelf-1 slot-10 1 }:line-interface]
signaling-mode = ss7-data-trunk
incoming-call-handling = internal-processing
```

Parameter	Setting for SS7 data trunks
signaling-mode	Must be set to ss7-data-trunk. A line configured as an SS7 data trunk carries no signaling, so it provides 24 (T1) or 32 (E1) 64-kbps channels. When you specify ss7-data-trunk signaling, the line is registered with the IPDC and the IPDC takes control of the line, telling the TAOS unit when to bring calls up or down.
incoming-call-handling	Must be set to internal-processing. Specifies how the TAOS unit processes incoming calls on this line. This value is the same for both H.323 and IPDC VoIP call processing.

Example of configuring a T3 profile

To configure lines of a T3 card as SS7 data trunks, you must first configure the T3 profile as in the following example:

```
admin> read t3 {1 1 1}
T3/{ shelf-1 slot-1 0 } read
admin> set enabled = yes
admin> set frame-type = m13
admin> set line-length = 0-225
admin> write
T3/{ shelf-1 slot-1 1 } written
```

After configuring the T3 line, configure the individual T1 lines that constitute the T3 line as explained in the next section.

Example of configuring a T1 data trunk

The following commands configure a T1 line as an SS7 data trunk, enabling IPDC to control the line:

```
admin> read t1 {1 1 7}
T1/{ shelf-1 slot-1 7 } read
admin> set line-interface enabled = yes
admin> set line-interface signaling-mode = ss7-data-trunk
admin> set line-interface incoming-call-handling = internal-processing
admin> write
T1/{ shelf-1 slot-1 7 } written
```


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Example of configuring an E1 data trunk

The following commands configure an E1 line as an SS7 data trunk, enabling IPDC, from the signaling gateway, to control the line:

```
admin> read e1 {1 10 1}
E1/{ shelf-1 slot-10 1 } read
admin> set line-interface enabled = yes
admin> set line-interface signaling-mode = ss7-data-trunk
admin> set line-interface incoming-call-handling = internal-processing
admin> write
E1/{ shelf-1 slot-10 1 } written
```

Configuring PRI Tunneling in IPDC (IPDC 0.15)

PRI tunneling allows ISDN layer 3 signaling to be tunneled to a signaling gateway. The external signaling gateway controls the T1/E1 PRI lines terminating on a MultiVoice Gateway.

When the TAOS unit acts an access gateway or trunking gateway, the T1/E1 PRI lines must be made visible to an external signaling gateway. The external signaling gateway uses IPDC for call control on these lines.

In this tunneled PRI signaling scheme, a MultiVoice Gateway handles layer 1 and layer 2 of PRI signaling. All layer 3 Q.931 messages on the D-channel are tunneled to the external gateway by means of an IPDC TUNL message. The bearer channels on the PRI lines are controlled by IPDC call setup and teardown messages.

Requirements

To support PRI tunneling, a TAOS unit requires the following:

- IPDC signaling must be enabled on the TAOS unit (that is, `xcom-ss7` must be set to `enabled` in the base profile).
- The IP address and TCP port to use as the IPDC interface to the SS7 signaling gateway must be specified. Typically, the primary and secondary address and port configurations point to the two Ethernet interfaces of the SS7 signaling gateway. Assign the appropriate IP address to the `primary-ip-address` parameter and the appropriate port number to the `primary-tcp-port` parameter in the `ss7-gateway` profile.
- The `signaling-mode` parameter must be set to `tunneled-pri-signaling` in the `line-interface` subprofile of an `t1` or `e1` profile. This value allows an external signaling gateway using IPDC to perform call control on T1/E1 lines terminating on a TAOS unit. The TAOS unit recognizes and responds to IDSN signaling, with local B channels controlled by an external signaling gateway. All layer 3 Q.931 messages are tunneled to the gateway configured in the `ss7-gateway` profile.

Note Only one signaling type can be used on a MultiVoice Gateway or channelized T1/E1 slot card. Only ISDN network terminated (NT) emulation for T1 and T3 lines that are connected to NI-2 and 5ESS/4ESS ISDN switch types are supported.

Reporting PRI tunneling status

The `status` command reports an active tunneled PRI trunk, using the symbol, `"i"`. This symbol identifies DS0 connections that use ISDN PRI with layer 3 tunneled signaling to the signaling gateway, as illustrated by the following:

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```
0 Connections, 0 Sessions| TNT22 Status
                         | Serial number: 9021340  Version: 9.0a0e0
                         |
                         |  Rx Pkt: 27763
                         |  Tx Pkt: 14688
                         |     col:     2
                         |
                         | 04/06/2000 18:29:42  Up: 0 days, 02:30:20
                         |-----------------------------------------
                         |T-PRI 1/01/01 LA i-------- -------- -------s
                         |                 ^
```

Using tunlpri command options

The `tunlpri` command is used to report the status of calls processed using tunneled PRI signaling. This command uses the following syntax:

admin> **tunlpri -s**

Using the -s option, the `tunlpri` command displays module statistics. To enable tunneled PRI diagnostics, use the following `diag` command to set the desired level for debugging tunneled PRI operations:
`diag tunlpri` *`level`*

Following are the values you can specify for *level*:

Debug level	Specifies
0x00	Diagnostic output is disabled. No debugging information is collected.
0x01	Report errors only. Collect only high level error information as errors occur.
0x02	Show basic debugging traces. Collect session logs.
0x04	Dump Tunnel messages. Collect the IPDC TUNL messages sent to and received from the SoftSwitch.
0x08	Show detailed debugging traces. Collect full session logs, including low-level processing information for tunneled PRI signaling.

The following example illustrates the output of the `tunlpri` command when debug level four (0x04) is specified:
admin>**tunlpri -s**

```
Tunneled PRI Module statistics:
 Interface initialized and ready:        Yes
    Current level of diagnostics:          15
Message count:
 Received from L2  :                     1068
 Sent to L2        :                     754
 Received from Tunl:                     945
 Sent to Tunl      :                     996
```


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```
Errors:
 Errors at startup:                    0
 Warnings:                             0
 Module usage errors:                  0
 NULL pointers:                        0
 Control Bus errors:                   72
 Buffer pools errors:                  0
 Protocol errors:                      0
 Total:                                72
APX6>
```

Modifications to the ss7nmi *command*

The ss7nmi debug-level command reports TUNL message statistics when entered as follows:

admin> **ss7nmi -m**

When the command is entered with the -m option, the results displayed include the number of tunneled PRI (TUNL) messages sent or received by the TAOS unit. The ss7nmi debug command includes the following options specifically for IPDC Tunneling debugging:

Options	Specifies
-m	Show TUNL message statistics
-mr	Reset TUNL message statistics
-n	Show active NLCBs (transactions)
-r [address]	Show the status of the SS7 circuit(s). When address is specified, show only status for the selected circuit.
-rc	Toggle, enable or disable, resource backtrace collection. By default, this option is disabled.
-rd [address]	Show detailed status of circuit(s). When address is specified, show only status for the selected circuit.
-s	Show SS7 interface statistics
-sr	Reset SS7 interface statistics

To enable tunneled PRI diagnostics, use the following diag command to set the desired level for tracing tunneled PRI messaging:
diag ss7nmi *level*

Debug level	Specifies
0x00	Diagnostic output is disabled. No debugging information is collected.
0x01	Report errors only. Collect only high level error information as errors occur.

Debug level	Specifies
0x02	Show signaling link states. Collect information on SS7 link-state changes.
0x04	Show NLCB/transaction states. Collect information on NCLB statuses and transactions state changes.
0x08	Show signaling semantics. Collect information on signaling types associated with each call.
0x10	Display contents of NMI packets. Collect information from network management information packets.
0x20	Show call control interface details. Collect information on the interface used to set up, monitor and tear down each call.
0x40	Show internal task events. Collect information on the low-level processes used for call control.
0x80	Show memory usage. Collect information on the memory allocated by the TAOS unit to process calls.
0x100	Show resource allocation details. Collect information on how TAOS unit resources are allocated for each call.
0x200	Show tunnel basic errors. Collect only high-level tunneling PRI error information as errors occur.
0x400	Show tunnel basic debug. Collect only high-level tunneling PRI debugging information for calls as they occur.
0x800	Dump Tunnel messages. Collect the IPDC TUNL messages sent to and received form the SoftSwitch.
0x1000	Show detailed debugging traces. Collect full session logs, including low-level processing information for tunneled PRI signaling.

The following example illustrates the output of the `ss7nmi -m` command, reporting the TUNL messaging statistics:

```
admin>ss7nmi -m
          IPDC message processing statistics:
               Message code        Received      Sent
               RCR    (0x0011):     152802          0
               ACR    (0x0012):          0     152802
               RCCP   (0x0013):     152847          0
               ACCP   (0x0014):          0     152847
               RMS    (0x0041):          1          0
               NMS    (0x0042):          0         24
               RLS    (0x0043):         28          0
               NLS    (0x0044):          0         29
               NCS    (0x0046):          0          7
```

```
TUNL  (0x007a):   611460   611480
RTE   (0x007d):      111        0
ARTE  (0x007e):        0      111
NSUP  (0x0081):        0        1
ASUP  (0x0082):        1        0
Data collection was started: [04/08/2002 17:24:01]
```

Configuring trunk signaling for H.323 VoIP networks

For every TAOS unit that operates in a H.323 VoIP network and connects to the PBX and PSTN, you must configure the T1 line settings to detect and respond to call progress signaling for H.323 VoIP calls. Set the following parameters, as shown with sample settings:

```
[in T1/{ shelf-1 slot-1 1 }:line-interface]
signaling-mode = inband
robbed-bit-mode = inc-w-200
default-call-type = voip
collect-incoming-digits = yes
t1-inter-digit-timeout = 6000
[in T1/{ shelf-1 slot-1 2 }:line-interface]
signaling-mode = isdn
default-call-type = digital

[in E1/{ shelf-1 slot-1 1 }:line-interface]
signaling-mode = r1-inband
robbed-bit-mode = inc-w-200
default-call-type = voip
number-complete = 7 digits
caller-id = get-caller-id
e1-inter-digit-timeout = 6000
[in E1/{ shelf-1 slot-1 3 }:line-interface]
signaling-mode = isdn
default-call-type = digital
```

Parameter	**Setting**
`signaling-mode`	Type of signal received from the T1 trunk. Set this parameter to: • `inband` for non-PRI T1 trunks using inband with robbed bit signaling. When using inband signaling (T1, MF R2), audible tones are used to transmit DNIS/ANI across the trunk. • `isdn` for T1/PRI trunks. When using ISDN signaling, DNIS/ANI are transmitted in the ISDN call setup message. • `dtmf-r2-signaling` for R2 signaling trunks. When using DTM R2 signaling, a DSP is allocated to detect DTMF tones for inbound and outbound calls (see "Enabling DTMF R2 signaling for E1 lines" on page 2-57 for details). • `inband-fgd-in-fgd-out` for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received and sent in FGD format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • `inband-fgd-in-fgc-out` for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received in FGD format and sent in FGC format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • `inband-fgc-in-fgc-out` for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received and sent in FGC format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details). • `inband-fgc-in-fgd-out` for Feature Group D (FGD) signaling on T1 inband trunks. Call signaling data is received in FGC format and sent in FGD format (see "Enabling and debugging Feature Group D signaling support for T1 lines" on page 2-58 for details).
`robbed-bit-mode`	Type of robbed bit-signaling received from an inband T1 trunk. Set this parameter to `inc-w-400` or `inc-w-200` for trunks supporting DNIS/ANI.

Parameter	Setting
`default-call-type`	Default calltype for incoming calls, when using non-PRI T1 trunks (inband with robbed-bit signaling), or when the TAOS unit is configured for single-staged dialing. Set the value of this parameter to `voip` if all calls received are processed as VoIP calls. For T1 configurations that don't provide DNIS (in-band), this method is required to map an incoming call as a VoIP call. If other call types (such as, modem calls) are received over this trunk, set the value to `digital`.
`collect-incoming-digits` (T1) `caller-id` (E1)	Enables/disables collection of the Dialed Number Identification Service (DNIS) string for the destination telephone number and Automatic Number Identification (ANI) string of the calling telephone number for an incoming call. For trunks supporting DNIS/ANI, set the value of this parameter to yes. This parameter is ignored when `signal-mode=isdn`. If two-stage dialing is being used with ISDN signaling, then the two-stage dialing parameter must be set to yes. **Note** If DNIS/ANI is present on the T1, you must set this parameter to yes, otherwise the H.323 signaling layer identifies the DNIS/ANI digits as part of the destination phone number dialed by the user. If DNIS/ANI is not desired, provision the PSTN switch/PBX not to send DNIS/ANI.
`t1-inter-digit-timeout` `e1-inter-digit-timeout`	How long the TAOS unit waits after receiving the last digit before declaring DNIS/ANI collection complete, when using inband signaling (T1, MF R2). The TAOS unit waits until this interval has elapsed to ensure it has received all audible tones used to transmit DNIS/ANI across the trunk. It may be set to a value between 100 and 6000msec. This parameter defaults to 3000msec. (3 seconds) and works with the Number-Complete parameter when time-out processing is implemented (for details, see "Enabling collections of variable length dial strings without EOP" on page 2-61). **Note** The `call-inter-digit-timeout` parameter in the `voip` profile is used to control collection of keypad generated DTMF entered by the caller, when using two-stage dialing. It has no effect on the collection of DNIS/ANI for T1 signaling.
`number-complete`	In the E1 profile, the condition that the MultiVoice Gateway uses to determine the length of the dial string. Up to 15 digits can be collected for R2 dial strings without waiting for end-of-pulse (EOP) signaling. Time-out processing can also be implemented by setting this parameter to `time-out` (see "Enabling collections of variable length dial strings without EOP" on page 2-61 for details).